Hardware implementation of blind source separation of speech signals using independent component analysis

Zahooruddin
Assistant Professor (zahooruddin79@gmail.com)
Farooq Alam Orakzai
Assistant Professor (farooqorakzai@hotmail.com)
Department of Electrical Engineering CIIT Wah Cantt, Pakistan

Abstract-- This paper consists of hardware implementation of fast ICA algorithms for mixed speech signals. During experiment we recorded the voice signals of three different people one by one with the help of a single mic. After mixing the incoming signals ICA was applied. The observed results were compared with the incoming voice signals in order to know accuracy of the algorithm. In a second experiment we designed an array of mic where different people were talking at the same time. After low pass filtering, amplification and A/D conversion data was recorded serially in a computer through a microcontroller. Again ICA was applied and results were observed by listening the voice through a headphone. The resultant voice signals were very clear.

Index Term-- ICA, mixed data, multiplexer, A/D converter, microcontroller, voice data

I. INTRODUCTION
The main objective of this research is hardware implementation of extracting speech signals from a set of mixed data. Fast ICA Algorithm [1] was implemented for blind source separation of mixed speech data. Fig.1 shows block diagram of the overall system. First of all voice signals were sensed by an array of mic and an analog multiplexer was used to convert the data into single channel. Conversion of data into single channel reduces our hardware and makes data processing easy. The signals were amplified after low pass filtering. Now analog data is converted into digital format and is stored in a computer with the help of a microcontroller [2]. During first experiment we recorded the voice signals of three people one by one with the help of a single mic which is shown in fig.4. After mixing of the incoming signals ICA was applied and the results were compared with the incoming voice signals in order to know accuracy of the algorithm. The results are summarized in table I. In a second experiment we designed an array of three mic where three people were talking at the same time. After low pass filtering [3] and Amplification [4], we used an analog multiplexer [5] to select simultaneously different incoming signals. Multiplexer address selections were controlled through a microcontroller. After analog to digital conversion data was recorded serially in a computer through a microcontroller. Actual mixed voice data was now available in a system which is shown in fig.2. ICA was applied and results were observed by listening the output voice signals through headphone which are shown in fig.3. The resultant voice signals were very clear.

A. FAST ICA ALGORITHM [6]
The Fast ICA algorithm involve the following steps,
1. Make the available mixed data zero mean
2. Whiten the data.
3. Choose an initial weight vector w of unit norm.
4. Let \( w_{new} = E\{m_i g(w^T m_i)\} - E\{m_i g'(w^T m_i)\}w \) This is the basic weight update equation, where \( g \) is the contrast function.
5. Let \( w_{new} = w_{new} / ||w_{new}|| \) This is the normalization step that makes the new \( w \) as unit norm, which will be updated at every iteration. Compare \( W_{new} \) with the old vector, if converged than move ahead, if not go to step 4.

![Fig. 1. Block diagram of the overall system](image1)

![Fig. 2. Mixed data](image2)
II. RESULTS AND DISCUSSION

Voice signals were sampled by 8 kHz frequency. We did the first experiment several times for different types of signals. The results of speech and mobile tones are given in Table I. Very small value of error standard deviation (STD) shows that the algorithm is very useful for this type of applications. [7]

This system has various applications like person identification in a crowd, biomedical signal extraction, information extraction from mixed data transmitted through a communication channel etc.

<table>
<thead>
<tr>
<th>S.No</th>
<th>Signals</th>
<th>Error STD</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Voice 1</td>
<td>0.0097</td>
</tr>
<tr>
<td></td>
<td>Voice 2</td>
<td>0.0011</td>
</tr>
<tr>
<td></td>
<td>Voice 3</td>
<td>0.0019</td>
</tr>
<tr>
<td>2</td>
<td>Mobile Tone 1</td>
<td>0.0257</td>
</tr>
<tr>
<td></td>
<td>Mobile Tone 2</td>
<td>0.0436</td>
</tr>
<tr>
<td></td>
<td>Mobile Tone 3</td>
<td>0.0177</td>
</tr>
</tbody>
</table>

REFERENCES